SIP Trunking

The emerging standard in enterprise voice.

What is SIP Trunking?

SIP Trunking is an innovative IP voice solution which provides businesses like yours with customized next-generation voice services. By replacing your costly PRIs and traditional analog lines, SIP Trunking simplifies your company's telephony infrastructure and saves you money.

Why SIP trunking is creating a buzz

- Complete your unified communications strategy and dramatically reduce telecom expenses:
  - Manage the same call capacity with fewer lines (or "channels")
  - Enjoy Capacity On Demand: accommodate any call volume by only paying for what you need, when you need it
  - Create a local presence and eliminate long distance charges by selecting local phone numbers in over 1000 cities across Canada
- Make changes to your account instantly thanks to sophisticated online management tools
- Create increased efficiency by pooling and seamlessly shifting capacity across multiple sites
- Bank on our redundant national voice network built on carrier-class technology

Is SIP trunking right for your business?

- Do you have an IP-ready PBX?
- Are you thinking about purchasing a new PBX?
- Do you have multiple office sites?
- Do you experience highly variable call volume?
- Do you ever experience issues with busy signals or insufficient lines?
- Are you seeking enterprise-level reliability and scalability?

Capacity On Demand

Most businesses that rely on a traditional voice infrastructure pay for up to 50% more lines than they need - just to be safe. With ThinkTel, you'll only pay for the lines that you really need because Capacity On Demand will always be available.

SIP Trunking delivers real cost savings

By switching to a customized SIP Trunking configuration, a company with 400 employees in 10 cities (40 employers per site) will experience a 36% reduction in telecom costs: a savings of over $10,000 per month!

100% Business continuity with Surecall

ThinkTel's exclusive Surecall feature allows you to specify a call forwarding number for each active DID. Should a SIP trunk go down due to ISP outages, network issues, or problems with your on-site PBX, Surecall will be there to ensure your inbound calls still reach the right person.

Try SIP Trunking...risk free!

Test drive SIP Trunking technology risk free by replacing some of your costly analogue lines and ISDN channels with SIP trunks, while your remaining ISDN and analog lines serve as back-up until you're ready to make a complete migration.

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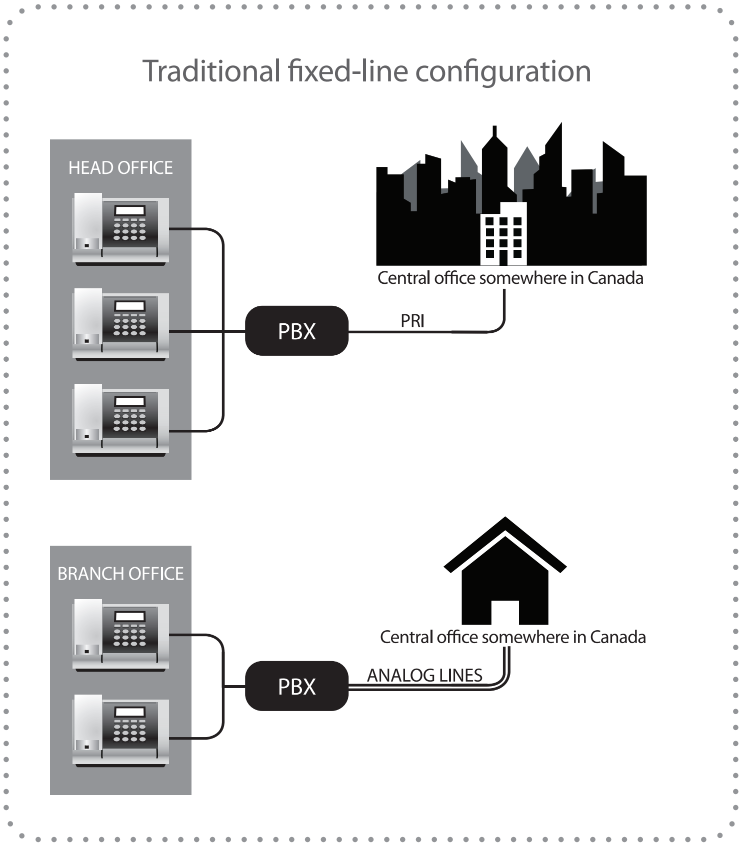
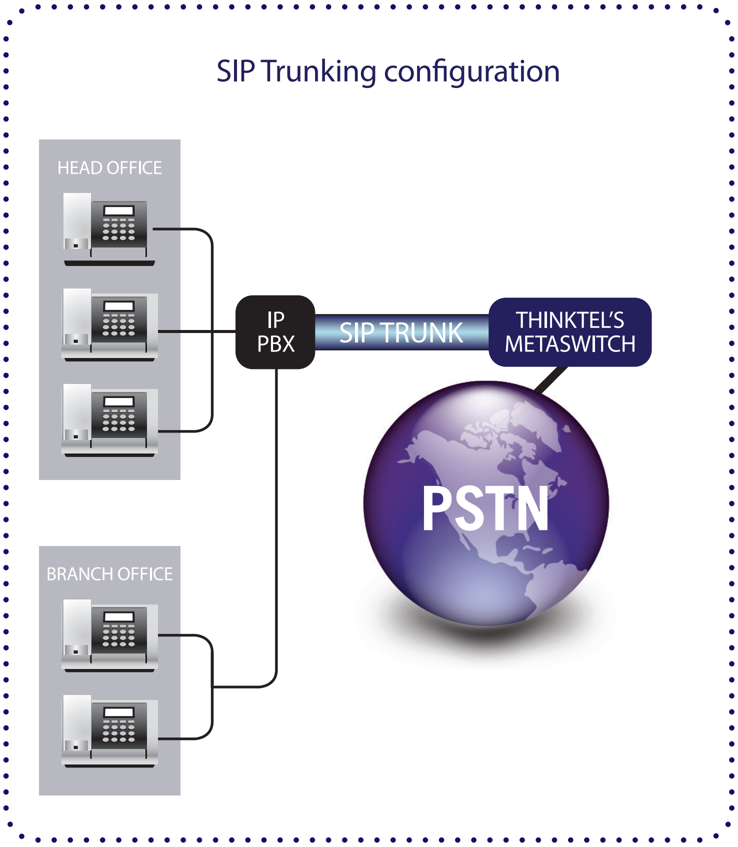
SIP Trunking

Streamline your network

Here's a high-level overview of how a SIP Trunking configuration differs from a traditional fixed-line setup





Network specifications and capabilities

- Up to 8 IP addresses per SIP Trunk for built-in redundancy
- Codec support for G.711, G.726, G.721, and G.729a/b
- QoS support via TOS bit
- Support for IPSec VPN
- T38 fax support
- Fully redundant class 4/5 core switching powered by MetaSwitches
- Directory listings for all DIDs
- 10 or 11 digit dialling (e.164)
- 15 character Caller ID and Caller Name
- Basic and VoIP V911 with user-managed location for each DID
- Private IP Cross Connection availability
- Certified with Cisco CallManager and UC 520/540/560PBX's
- Certified directly with Microsoft OCS R2 and Lync

What you need to successfully deploy a SIP trunk:

- Sufficient bandwidth to handle your voice needs
- An IP-PBX with a SIP-enabled trunk side
- A SIP-ready enterprise edge device

For additional information, please contact us.
We'll be happy to help.

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